Kerio Operator

Step-by-Step

Kerio Technologies

This guide provides detailed description on *Kerio Operator*, version *1.0*. All additional modifications and updates reserved.

For current versions of the product and related manuals, check http://www.kerio.com/operator/download/.

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Contents

Chapter 1

Introduction

This document is a simple guide focused on description of *Kerio Operator* configuration, its first installation and startup in the network. To make the guide as comprehensible as possible, let us define an exemplary implementation of *Kerio Operator* (see figure 1.1):

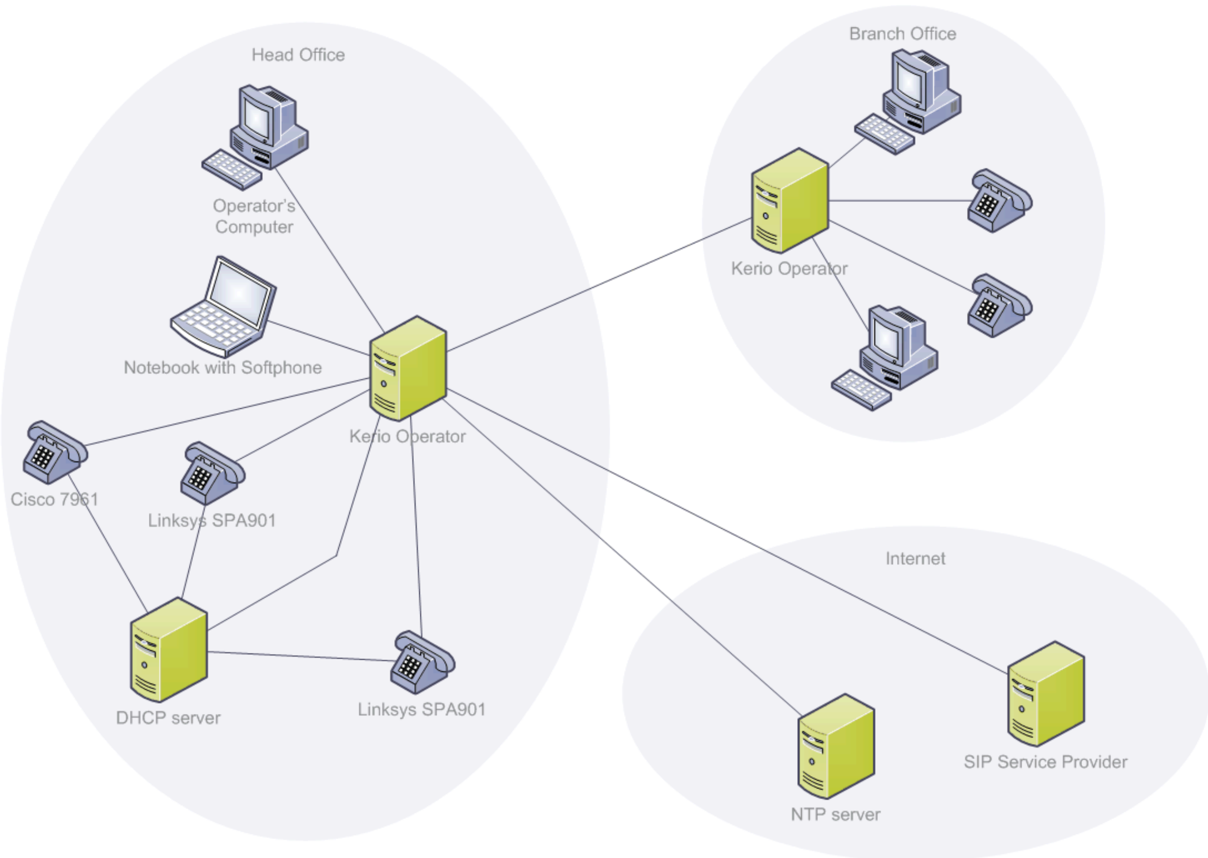

Figure 1.1 Exemplary implementation scheme

1. *Kerio Operator* is installed in a local network behind a firewall.
2. *Kerio Operator* will install and startup in a local network on computer `operator.company.com`.
3. *Kerio Operator* will be connected to a phone network via a SIP provider. An interface for incoming and outgoing calls communicating with the SIP provider will be configured.
4. Originally, the company used one number `555 0100` which was mapped to the operator's extension. After acquiring *Kerio Operator*, the company decided to obtain a complete

range of numbers with one valid digit 555 0120 — 555 0129 from a different provider. The original number will not be canceled because the customers are accustomed to it. The dial plan can be created by mapping the original number `555 0100` to the operator (company receptionist) together with one of the numbers from the new range (for example `555 0120`). The operator will transfer calls to the internal phone network manually. The rest of the numbers will be used for direct 1:1 mapping for sales department and company management.

5. Several local extensions and user accounts will be created for users and an operator. Some users will use software phones (softphones) on their computers.

6. Automatic hardware phone provisiong will be set — make sure DHCP is working.

7. *Kerio Operator* is connected to an NTP server (for time synchronization) to ensure correct time for logging information.

8. Since we have a main office and a branch in the exemplary implementation, we will create a conference call for several participants to ensure that the management of both offices can arrange phone meetings. We will limit the number of conference users to four and it will be protected by a PIN number.

9. The exemplary company has a technical support. We will create a call queue where *Kerio Operator* will automatically distribute customer calls among the specified company employees.

10. Auto attendant script will be configured in *Kerio Operator*. This function allows you to create a voice menu which is played to callers. Callers will navigate through the menu by pressing the phone buttons.

11. We will configure a software phone manually.

12. We will connect to *Kerio MyPhone*.

The exemplary configuration can be easily customized. For detailed information on setting of individual features of *Kerio Operator*, refer to *Kerio Operator, Administrator's Guide*. The whole document can be downloaded from the *Kerio Technologies* website at http://www.kerio.com/operator/manual.

1.1 Before we start

To configure your network, you need:

- *Kerio Operator* — a PBX.
- Hardware and/or software phones for users who will use the PBX.
- For outgoing calls, you need to have a contract with a commercial VoIP provider.
- You have to get a phone number or a SIP trunk with an interval of phone numbers that will be mapped to your internal phone network.

- Your provider needs to give you an IP address or name and port (usually 5060) of a SIP server for *Kerio Operator* to connect to.
- Get an authentication name and password for the SIP server from the provider. Many providers use a phone number as the authentication name.
- Ask your provider whether *Kerio Operator* needs to register to a registrar or proxy server before the first connection to a SIP server. The registration is usually required for accounts with one number. If you have a SIP trunk with an interval of phone numbers, IP-based authentication is usually used.

Chapter 2

Installation

Kerio Operator is available as an ISO image of the installation disk. You can burn the installation disk and install it on a computer with the following minimum requirements:

- CPU Celeron 1 GHz,
- 512 MB RAM,
- 8 GB free disk space,
- Ethernet card.

No operating system is required to be installed on the computer. Any existing operating system will be removed during the *Kerio Operator* installation.

Chapter 3

Server configuration

3.1 Accessing administration interface

The *Kerio Operator* PBX is configured through the *Kerio Web Administration* interface. To connect to this interface, use URL following this pattern: `https://IP.address/admin`.

The URL will be for example: `https://192.168.10.1/admin`.

If the URL is correct, a dialog is displayed requiring a password for the administrator's account (`Admin` account). Enter and confirm the password and create an administrator's account with the full *Kerio Operator* administration rights. Once this dialog is confirmed, a login page appears requiring the same username and password before you can continue.

> ***Warning:***
> We recommend using a strong password.

3.2 Creating local user accounts

When you login to *Kerio Operator* for the first time, create a user account for an operator and assign them a phone extension. Similarly, you can create accounts for other users, if you do not wish to map them from a directory service (see the Kerio Operator, Administrator's Guide manual).

New user accounts can be defined in section *Configuration* → *Users*:

1. Click on *Add*.
2. Enter at least username and password.
3. Go to the *Extensions* tab and click on *Add*.
4. In the *Select Extensions* dialog, click the *Add* button.
5. In the *Extension number* field, enter an extension which has not been used yet. In our example, let us select extension 10.
6. Click OK to confirm settings.

3.3 Setting hardware phone provisioning

New hardware phones, which *Kerio Operator* can configure automatically, were bought for the new phone network. *Kerio Operator* in its current version supports the following phones:

- Cisco 7940 and 7960 with the SIP firmware in version 3 and higher,
- Linksys SPA942, SPA962, SPA922, SPA901, PAP2T with firmware in version 5 and higher.
- Snom 300, 360, 820, m3 and MeetingPoint.

Apart from the settings in the administration interface, hardware phone provisioning also requires a running DHCP server in the network. DHCP server must support parameter 66 (TFTP server address)[1]:

1. In parameter 66, set the name or IP address of *Kerio Operator*.

2. Go to *Kerio Operator* administration interface to section *Configuration → Provisioned Phones*.

3. Check whether the *Enable provisioning* option is enabled.

4. Enter the number (extension) which will start automatic phones provisioning. If the first number is set to 10, the next one will be 11 and so on. If a number is already in use (if the extension was created manually), it will be skipped.

5. Enter a password which hardware phones employing provisioning will use to authenticate.

Detailed information on the whole provisioning process can be found in manual Kerio Operator, Administrator's Guide.

3.4 Dial out setting

For outgoing calls and calls to the company branch, it is necessary to set a call interface.

Connection to branch office

To set a connection with the branch office, go to section *Configuration → Call Routing*:

1. Click on *Add a SIP interface*. This opens the configuration wizard.

2. In the first step, enter the name for the interface (for example `branch`). The name cannot contain spaces, national and special characters and must be unique.

3. Select the *Link to another PBX* option and click on *Next*.

[1] DHCP server integrated in *Kerio Control* supports parameter 66.

4. Enter a prefix for outgoing calls. The prefix tells *Kerio Operator* to which interface the call should be directed. For example, set number 5 as the prefix.

5. In the next dialog window, enter the address and port of the branch server in the *Hostname or IP address* and *Port* fields.

For detailed information on setting an interface between a head office and a branch, refer to the appropriate chapter in manual Kerio Operator, Administrator's Guide.

Connection to provider

To set a connection with an outer network, we need to create an interface to a SIP provider. According to the exemplary implementation, the provider has provided us with a range of numbers. One number was assigned earlier. Now, we create two interfaces. One for number `555 0100` and the other for the range of numbers.

1. In the administration interface, go to section *Configuration* → *Call Routing*.

2. Click on *Add a SIP interface*. This opens a dialog with a configuration wizard.

3. In the first step, enter the name for the interface (for example `provider1`). The name cannot contain spaces, national and special characters and must be unique.

4. Select the *New provider* option and enter the phone number (or range of numbers) assigned by the provider. Click on *Next*.

5. Select an extension for the operator (10, in our case) who will transfer all the calls from outer network to the number assigned by the provider to all the extensions created in *Kerio Operator*.

6. In the *Prefix to dial out* field, enter a prefix other then 5, which we use for calls to the branch office.

   Using prefixes is subject to local customs in each country (for example, in the USA, prefix 9 is usually used, in some European countries, prefix 0 is used). We will use prefix 8 for the original separate number and we will later set the common prefix 0 for the range of numbers.

7. In the next dialog window, enter the address and port of the SIP server in *Hostname or IP address* and *Port* fields (this information is provided by the provider).

8. The providers often require an authentication with a username and password (the username is usually the assigned phone number). If you have the login data available, enter them into *Username* and *Password*.

9. Before the first *Kerio Operator* connection, the provider also usually requires a registration with a registrar or proxy server. If this is the case, check the *Must register with the Registrar or Proxy* option.

We will configure the newly purchased SIP trunk with the following steps:

1. In the administration interface, stay in section *Configuration* → *Call Routing* and click on *Add a SIP interface....*
2. In the *Add SIP interface* dialog, enter the name for the interface (for example `provider2`). The name cannot contain spaces, national and special characters and must be unique.
3. Select the *New provider* option and enter the range of numbers assigned by the provider. See the dialog for an example of setting a range of numbers. In the exemplary implementation, we have a range of 10 numbers available, so we enter the number as 555 012X. Click on *Next*.
4. Proceed further according to the previous instructions (from step 5). Only set 0 as the prefix.

Kerio Operator creates a new SIP interfaces together with rules for incoming and outgoing calls.

Rules for call routing need to be adjusted:

1. In section *Configuration* → *Call Routing* in table *Interfaces and routing of incoming calls*, click on the line under the new SIP interface (provider2).
2. This opens the *Edit Incoming Call* dialog. Strip the called number from left to leave only one digit (the last one) because we have been assigned a range of numbers with only one valid digit. Since we use two-digit extensions, add digit 1 as a prefix to the called number.

   Strip number 555 0120 to 0 and add 1. Thus we create the required operator's extension 10. Mapping of a SIP trunk with an interval of numbers is in table 3.1

Interval of assigned numbers	Mapped extensions
555 0120	10
555 0121	11
555 0122	12
555 0123	13
555 0124	14
555 0125	15
555 0126	16
555 0127	17
555 0128	18
555 0129	19

Table 3.1 Direct mapping of SIP trunk

Rules for routing outgoing calls are also displayed in section *Configuration → Call Routing*.

For detailed information on setting an interface for incoming and outgoing calls, refer to the appropriate chapter in manual Kerio Operator, Administrator's Guide.

Route backup

Now we create a simple backup. If one of the providers is unavailable, the backup enables outgoing calls via the other provider.

1. In the administration interface, go to section *Configuration → Call Routing*.
2. In the *Routing of outgoing calls* section, click on the interface for prefix 8.
3. On the *General* tab, add provider2 to table *Use the following external interfaces*.
4. Click on the interface for prefix 0 and on the *General* tab, add provider1 to table *Use the following external interfaces*.

The backup is now set for both interfaces. If one of the connections is interrupted, the other interface will be used automatically for outgoing calls.

3.5 Creating call queue

Call queue can be defined in section *Configuration → Call Queues*:

1. Click on *Add* and enter the name of the queue on the *General* tab.
2. In the *Queue extension* field, enter an extension which is not used by any other user.
3. In the *Queue strategy* menu, select the *Round robin with memory* strategy — employees operating this queue (let us call them agents) will receive calls always in the same sequence.
4. Go to *Agents* tab and check the *Allow dynamic agent login/logout* option. Dynamic login and logout means that every agent may logout from the queue for example before lunch break and login again after their return.
5. Dynamic login and logout requires a code which the agents will use to login and to logout. Enter, for example, code 611 for login and code 622 for logout from the call queue. Codes must be unique in *Kerio Operator* — no extension with the same digits can be defined.

For detailed guidelines for setting call queues, refer to Kerio Operator, Administrator's Guide.

3.6 Creating conference call

Conference call (a concurrent call of more than two users) is created in section *Configuration* → *Conferences*:

1. Click on *Add* and enter a conference name in the dialog.
2. In the *Conference extension* field, enter an extension which is not used by any other user.
3. Limit the number of participants to 4 (we recommend to limit the conferences if the CPU utilization on your server is high).
4. Set a PIN number as a protection against tapping. This PIN number will also be used for access to the conference call.

To access the conference call, users dial the conference extension and are invited to enter the PIN. Once the PIN code is entered, they may communicate.

For detailed guidelines for setting conference calls, refer to Kerio Operator, Administrator's Guide.

3.7 Setting the Auto Attendant Script

Before the menu is selected, it is necessary to prepare (record) voice recordings which will constitute the menu. The recordings must be in GSM or WAV format. The recording may go as follows:

Hello! Welcome to XY company. For customer support, press `1`, for sales department, press `2`, for marketing department, press `3`. If you wish to speak with the operator, press `4`. If you hold on, you will be connected to the operator. The recording's name is `menu1.wav`.

New automatic scripts can be created under *Configuration* → *Auto attendant scripts*, by clicking on *Add*:

1. In the script number field, enter an extension which will launch the script.
2. In the *Description* field, enter text which will identify the script's content. In our example, it is `Main menu`.
3. Click on *Edit*. It opens the *Edit menu* dialog window for the `Main menu` script.
4. Click on *Select* next to *Announcement*. In the *Select audio file* dialog, select previously prepared recording as the *Announcement*.
5. Now complete the table. Click on *Add* and enter `1` in the *Key* column. In the *Action* column, select *Dial extension number*, and in the third column, enter the extension number to be dialed after pressing button `1`. Complete the remainder of the table similarly (for a complete example, see table 3.2).
6. In the *Default action* menu, select *Dial extension number* and enter extension 10. Thus the user will be connected to the operator unless they select an option from the menu.

Key	Action		Announcement
1	Dial extension number	11	
2	Dial extension number	12	
3	Dial extension number	13	
4	Dial extension number	10	

Table 3.2 Auto attendant script example

For detailed guidelines for setting auto attendant scripts, refer to Kerio Operator, Administrator's Guide.

Chapter 4

Client configuration

4.1 Connection to Kerio MyPhone

Kerio MyPhone is a web interface which allows you to change settings of your phone account in *Kerio Operator* from any place using a web browser. You can control your voicemail and the Find me function.

Login to *Kerio MyPhone* as follows:

To access the HTTP service using a web browser, insert the IP address (or the name if it is contained in DNS) of the computer where *Kerio MyPhone* is running. A protocol has to be specified in the URL — either HTTP for non-secured access or HTTPS for SSL-encrypted access. The URL will be as follows: `http://operator.company.com` or `https://operator.company.com`.

It is recommended to use the HTTPS protocol for remote access to the service (simple HTTP can be tapped and the user login data can be misused). By default, the *HTTP* and *HTTPS* services use the standard ports (80 and 443).

If a correct URL is entered, an authentication page is opened requiring a username and a password. Click on *Connect* to connect to the account.

4.2 Configuring software phones

To configure a software phone, have the following information ready:

- IP address or DNS name of the computer with *Kerio Operator*,
- password for the extension in *Kerio Operator* (password for the user account),
- assigned extension.

There are various types of software phones. Please, respect differences in their configurations:

1. Install a software phone.
2. Go to account settings.
3. Enter the extension number in the *Username*, *Extension* or *Phone number* field.
4. Enter password in the *Password* field.

5. Enter the DNS name or IP address of *Kerio Operator* in the *Domain* or *Registrator* field.

6. If the configuration contains the *Authentication name* field, enter the username of the account in *Kerio Operator*.

For detailed information on software phone configuration, refer to Kerio Operator, User's Guide The manual contains specific examples of configuration of X-Lite, Ekiga and SJphone phones.

Appendix A

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